AMENDMENTS TO THE CLAIMS

A detailed listing of all claims that are, or were, in the present application, irrespective of whether the claim(s) remains under examination in the application are presented below. The claims are presented in ascending order and each includes one status identifier. Those claims not cancelled or withdrawn but amended by the current amendment utilize the following notations for amendment: 1. deleted matter is shown by strikethrough for six or more characters and double brackets for five or less characters; and 2. added matter is shown by underlining.

1. (Currently Amended) A method for processing an electric sound signal:

wherein an electric sound signal on the right and an electric sound signal on the left are processed to produce a processed electric sound signal on the right and a processed electric sound signal on the left including the steps of:

simulating ~~the production of~~ a first processed electric ~~sound~~ signal ~~on the right from the electric sound signal on the right~~ corresponding to the detection by a right microphone of a right sound signal diffused in a reflective environment;

simulating ~~the production of~~ a second processed electric ~~sound~~ signal ~~on the right from the electric sound signal on the left,~~ corresponding to the detection by a left microphone of the right sound signal diffused in said reflective environment;

simulating ~~the production of~~ a third processed electric ~~sound~~ signal ~~on the left from the electric sound signal on the left~~ corresponding to the detection by a right microphone of a left sound signal diffused in said reflective environment;

simulating ~~the production of~~ a fourth processed electric ~~sound~~ signal ~~on the left from the electric sound signal on the right~~ corresponding to the detection by a left microphone of the left sound signal diffused in said reflective environment; ~~and~~

and wherein the right sound signal and the left sound signal correspond to the diffusion of the electric sound signal on the right and the electric sound signal on the left by a right speaker and a left speaker respectively,

the processed electric signal on the right is a combination of the first and third processed electric signal, and

the processed electric signal on the left is a combination of the second and fourth processed electric signal.

~~diffusing a sound corresponding to these four processed electric sound signals.~~

2. (Previously Presented) The method according to claim 1, wherein the simulating includes:

producing a white acoustic sound signal on the right is with an acoustic diffusion system, from a white noise electric signal;

detecting with an acoustic detector a corresponding acoustic signal received in the form of a modified white received electric sound signal on the right and a modified white electric sound signal on the left corresponding to the reception of the white acoustic sound signal on the right;

producing a frequency spectrum on the right corresponding to a white noise electric signal on the right, and two received frequency spectrums, respectively corresponding to the modified white received electric sound signal on the right and to the modified white received electric sound signal on the left;

producing a first set of coefficients from frequency filters from the frequency spectrum on the right and from the frequency spectrum of the modified white received electric sound signal on the right;

producing a second set of coefficients from frequency filters from the frequency spectrum on the right and from the frequency spectrum of the modified white received electric sound signal on the left;

producing a white acoustic sound signal on the left with an acoustic diffusion system, from a white noise electric signal;

detecting a corresponding acoustic signal received in the form of a modified white received electric sound signal on the left and a modified white electric sound signal on the right corresponding to the reception of the white acoustic sound signal on the left with an acoustic detector;

producing a frequency spectrum on the left corresponding to a white noise electric signal on the left, and two received frequency spectrums, respectively corresponding to the modified white received electric sound signal on the left and to the modified white received electric sound signal on the right;

producing a third set of coefficients from frequency filters from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the left;

producing a fourth set of coefficients from frequency filters from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the right, said

four sets of coefficients forming a quadrille of coefficient sets; and

filtering the electric sound signals on the right and left with frequency filters whose parameters are given by said quadrille.

3. (Previously Presented) The method according to claim 2, wherein:

the sets of coefficients are produced from the two spectrums by a component to component complex division of complex points from these components in each of these spectrums.

4. (Previously Presented) The method according to claim 2 wherein said diffusion includes the steps of

producing the coefficients from four temporal filters from coefficients of the first, second, third and fourth frequency filters respectively.

5. (Previously Presented) The method according to claim 4, wherein

the coefficients of temporal filters are modified by an operation including at least one of the steps of:

normalizing temporal filters of a quadrille, on the maximum of the direct field or on quadratic average of the diffuse field;

temporal resetting of the temporal filters with relation to each other;

providing a time lag of samples from a temporal filter;

masking of some samples from the temporal filter;

alteration of amplitudes from certain samples from a temporal filter.

6. (Previously Presented) The method according to claim 4 wherein

the coefficients from a temporal filter those whose rank is greater than a given rank are eliminated and where

in the coefficients from a temporal filter those whose value is lower than a threshold are eliminated.

7. (Previously Presented) The method according to claim 2 wherein

quadrilles of sets of coefficients are produced for different configurations of the acoustic diffusion system and or for different rooms in which the acoustic diffusion system is placed for the production of coefficients.

8. (Previously Presented) The method according to claim 7, wherein

one of the configurations is a configuration in cone of confusion.

9. (Previously Presented) The method according to claim 1 wherein, to diffuse,

the electric sound signals processed by the filters are combined with the original unprocessed electric sound signals,

and a combined electric sound signal on the right and a combined electric sound signal on the left are obtained.

10. (Previously Presented) The method according to claim 9, wherein, to combine,

a time lag is introduced between the acoustic electric sound signals processed by the filters and the original unprocessed electric sound signals.

11. (Previously Presented) The method according to claim 9 wherein

combined electric sound signals on the right and left are filtered on given frequency bands and,

a delay is introduced in each of these frequency bands.

12. (Previously Presented) The method according to claim 11, wherein

combined electric sound signals on the right and left are filtered by using a high-pass filter, and

high-frequency electric sound signals are obtained,

combined electric sound signals on the right and left are filtered by using a low-pass filter, and

low-frequency electric sound signals are obtained.

13. (Previously Presented) The method according to claim 12, wherein

a first delay is introduced in the low-frequency electric sound signals and

a second delay is introduced in the high-frequency electric sound signals.

14. (Previously Presented) The method according to claim 13, wherein

the first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the

first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the left, and

the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the left.

15. (Previously Presented) The method according to claim 1 wherein, to filter,

a signal transform of an electric sound signal is performed and a transformed signal is obtained,

the transformed signal is multiplied by the filtering coefficients and a multiplied signal is obtained,

the multiplied signal is transformed by an inverse transform, and

the filtering coefficients are coefficients of finite impulse response filters.

16. (Previously Presented) The method according to claim 15, wherein, to perform the transform

a frame of the electric sound symbol is divided into N blocks,

the transform of each of the blocks is performed,

the filtering coefficients are divided into N packets of coefficients,

the N blocks of input data are multiplied two by two by the N packets of filter coefficients, and

the multiplied blocks are added to obtain the multiplied signal.

17. (Previously Presented) The method according to claim 16, wherein to divide the frame and to calculate the transform,

the transform of each of the N blocks is calculated successively, and

the transformed blocks are transmitted to a delay line at N outputs.

18. (Previously Presented) The method according to claim 16 wherein, to divide the frame into N blocks,

an electric sound signal is stored in a circular buffer memory with capacity proportional to the nth of the frame of the electric sound signal.

19. (Previously Presented) The method according to claim 16 wherein,

to divide a frame of the signal into N blocks, double blocks are formed that are overlayed on each other by half,

the transform of each of the double blocks is performed,

the N packets of coefficients are completed by the constant samples to obtain double packets,

each of the N double blocks are multiplied by one of the N double packets and multiplied double blocks are obtained, and

the multiplied blocks are extracted from the multiplied double blocks.

20. (Currently Amended) The method according to claim 1 wherein, to simulate,

an artificial head that comprises two acoustic detectors ~~(68,69)~~ is placed in a median axis of two acoustic diffusion systems ~~(65,66)~~,

an electric signal in the form of a Dirac comb is applied simultaneously as input to the two acoustic diffusion systems, and

these direct fields and these crossed fields received by the acoustic detectors are aligned two by two by varying the position of the artificial head.

21. (Previously Presented) The method according to claim 1 wherein, to diffuse,

equalization functions are incorporated in the cells situated upstream from the Fourier transform cells.

22. (Previously Presented) The method according to claim 21, wherein

the frequency components of four frequency filters obtained from four modified temporal filters are adjusted independently.

23. (Previously Presented) The method according to claim 1 wherein, to diffuse,

the phase and/or the amplitude of the temporal filter coefficients are modified along all or part of the impulse response.

24. (Previously Presented) The method according to claim 15, wherein, to perform the transform,

the filtering temporal coefficients are divided into Q slots (HDD1-HDD4) of coefficients with progressive length M, 2M, 4M,...(2^(Q-1))M points,

the transform of each of these slots is performed and transformed slots are obtained,

a frame of the electric sound signal is divided into blocks (x1-x8) with a length of M points,

the transform of each of these blocks is performed and transformed blocks are obtained, and

the transformed blocks are multiplied by the transformed slots and corresponding multiplied blocks are obtained by inverse transformation to the blocks of signals that half-overlap each other two by two in time.

25. (Currently Amended) The method according to claim 24 wherein, to perform the inverse transformations of multiplied blocks,

a first multiplied block with a length of 2P x M points, a temporal block corresponding in time to this first multiplied block, a second multiplied block corresponding in time to a second temporal block are modulated , this first and second temporal block are overlayed by half in time, and

a modulated block with a length of 2P x M points is obtained, then

this modulated block with a length of 2P x M points is added to the second block, and

a combined block [[(621)]] with a length of 2P x M points is obtained.

26. (Previously Presented) The method according to claim 25, wherein, to modulate,

the odd components of a multiplied block with a length of 2M points wherein the block corresponding to it in time is overlayed with another is multiplied by –1, and the even components are multiplied by +1.

27. (Previously Presented) The method according to claim 25 wherein, to perform the inverse transformations of multiplied blocks with a length of 2M points,

the even components of the combined block with a length of 2P x M points are selected, and

an even block with a length of 2(P-1) x M points is obtained

this even block is multiplied by 1/2 and the result of this multiplication is added to an auxiliary multiplied block with a length of 2(P-1) x M points, and

a compensation block is obtained.

28. (Previously Presented) The method according to claim 25 wherein to perform the inverse transformations of multiplied blocks with a size of (2P)M,

the odd components of the combined block with a size of 2P x M points are selected, and

an odd block with a length of 2(P-1) x M points is obtained,

an inverse transform of this odd block with a length of (2(P-1))M points is performed, and

an odd inversed block is obtained that is situated in the temporal domain, then

this odd inversed block is multiplied by a complex coefficient conjugated from a complex coefficient W(n), and

an odd normalized inversed block with a length of 2(P-1) x M points is obtained.